Please amend the paragraph on page 10, lines 6 to 18, as follows:

sub C2 B2

As is obvious from the above aspects, according to the present invention, the mode is discriminated on the basis of the past quantized gain of the adaptive codebook. If a predetermined mode is discriminated, combinations of code vectors stored in the codebook, which [is] <u>are</u> used to collectively quantize the amplitude or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions are searched to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. With this arrangement, even if the bit rate is low, a background noise portion can be properly coded with a relatively small [amount] calculation amount.

Please amend the paragraph on page 12, lines 2 to 18, as follows:

sub C3 B3

Several embodiments of the present invention will be described below with reference to the accompanying drawings. In a speech coding apparatus according to an embodiment of the present invention, a mode discrimination circuit (370 in Fig. 1) discriminates the mode on the basis of the past quantized gain of an adaptive codebook. When a predetermined mode is discriminated, a sound source quantization circuit (350 in Fig. 1) searches combinations of code vectors stored in a codebook (351 or 352 in Fig. 1), which is used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. A gain quantization circuit ([365] <u>366</u> in Fig. 1) quantizes gains by using a gain codebook (380 in Fig. 1).

Please amend the paragraph beginning on page 17, line 19, and continuing to page 18, line 5, as follows:

sub C4 B4 cont

For example, linear predictive coefficients calculated for the second and fourth subframes based on the Burg method are transformed into LSP paramete3rs

sub-C4) B4

whereas LSP parameters for the first and third subframes are determined by linear interpolation, and the LSP parameters of the first and third subframes are inversely transformed into linear predictive coefficients. Then, the linear predictive coefficients α il (i=1, . . ., 10, [1] l=1, . . .,5) of the first to fourth subframes are output to a perceptual weighting circuit 230. The LSP parameters of the fourth subframe are output to the spectrum parameter quantization circuit 210.

Please amend the paragraph on page 18, lines 6 to 15, as follows:

sub-C5) B5

The spectrum parameter quantization circuit 210 efficiently quantizes the LSP parameters of a predetermined subframe from the spectrum parameters and outputs a quantization value which minimizes the distortion given by:

$$D_j = \sum_{i=1}^{p} W(i)[LSP(i) - QLSP(i)_j]^2 \qquad ...(1)$$

where LSP(i), $QLSP(i)_j$, and W(i) are the LSP [parameter] <u>parameters</u> of the ith-order before quantization, the jth result after the quantization, and the weighting coefficient, respectively.

Please amend the paragraph on page 20, lines 6 to 15, as follows:

sub-C6) B6

The LSP parameters of the first to third subframes reconstructed in such a manner as described above and the quantization LSP parameters of the fourth subframe are transformed into linear predictive coefficients α il (i=1, . . ., 10, [1] <u>l</u>=1, . . .,5) for each subframe, and the linear predictive coefficients are output to the impulse response calculation circuit 310. Furthermore, an index representing the code vector of the quantization LSP parameters of the fourth subframe is output to a multiplexer 400.

Please amend the paragraph on page 20, lines 16 to 22, as follows:

sub C7) B7

The perceptual weighting circuit 230 receives the linear predictive coefficients α il (i=1, . . ., 10, [1] <u>l</u>=1, . . .,5) before quantization for each subframe

sub C7 cont. B7

from the spectrum parameter calculation circuit 200, performs perceptual weighting for the speech signal of the subframe on the basis of the method described in reference 1 and outputs a resultant preceptual weighting signal.

Please amend the paragraph beginning on page 22, line 13, and continuing to page 24, line 2, as follows:

B8

The adaptive codebook circuit 500 receives a sound source signal v(n) in the past from a gain quantization circuit 366, receives the output signal $x'_w(n)$ from the subtractor 235 and the impulse responses $h_w(n)$ from the impulse response calculation circuit 310. Then, the adaptive codebook circuit 500 calculates a delay [DT] D_T corresponding to pitch, which minimizes the distortion given by:

sub C8

$$D_T = \sum_{n=0}^{N-1} x'^2_w(n) - \frac{\left[\sum_{n=0}^{N-1} x'_w(n)\, y_w(n-T)\right]^2}{\left[\sum_{n-0}^{N-1} y^2_w(n-T)\right]} \qquad \ldots(7)$$

$$\text{for } y_w(n-T) = v(n-T)*h_w(n) \qquad \ldots(8)$$

and outputs an index representing the delay to the multiplexer 400[.], where the symbol * signifies a convolution calculation.

Please amend the paragraph on page 24, lines 10 to 16, as follows:

sub C9 B9

For a voiced sound, a B-bit amplitude codebook or polarity codebook is used to collectively quantize the amplitudes of pules in units of M pulses. A case wherein the polarity codebook is used will be described below. This polarity codebook is stored in a codebook 351 for a voiced sound, and is [store din] stored in a codebook 352 for an unvoiced sound.

Please amend the paragraph beginning on page 24, line 24, and continuing to page 25, line 4, as follows:

Equation (11) can be minimized by obtaining a combination of an amplitude code vector k and a position [mi] $\underline{m_i}$ which maximizes $D_{(k,\ i)}$ given by:

B10

$$D_{(k,\ j)} = \frac{\left[\sum_{n=0}^{N-1} e_w(n)\ s_{wk}(m_i)\right]^2}{\sum_{n=0}^{N-1} s_{wk}^2(m_i)} \qquad \ldots(12)$$

sub C10

where [s_{wk}(mi)] $\underline{s_{wk}(m_i)}$ is calculated according to equation (5) above.

Please amend the paragraph on page 32, lines 7 to 14, as follows:

sub C11

Referring to Fig. [15] $\underline{5}$, in the fifth embodiment of the present invention, a demultiplexer section 510 demultiplexes a code sequence input through an input terminal 500 into a spectrum parameter, an adaptive codebook delay, an adaptive codebook vector, a sound source gain, an amblitude or polarity code vector as sound source information, and a code representing a pulse position, and outputs them.

<u>In the Claims</u>:

Please amend claims 6 and 7 as follows. A clean copy of the amended claims is attached.

sub C12

B11

Claim 6 (Twice Amended). A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

Sub C12

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to [poutout] <u>output</u> a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including[:]

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codeboook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized

sub C12 B11 concl

sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

B12

sub C12 cont

Claim 7 (Twice Amended). A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of [a] <u>an</u> adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section,

said sound source quantization section [for] outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output

sub C12 cont B17

from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

B17

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes spectrum parameters and reproduces a speech signal by filtering the sound source signal.

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sub C12 cont B27

Claim 8 (Amended). A speech coding apparatus comprising:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal; and

mode discrimination means for receiving a past quantized adaptive codebook gain and [performs] performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

sound source quantization means for quantizing a sound source

B13

Sub C12 cont

signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally [shifting] shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

gain quantization means for quantizing a gain by using a gain codebook; and

multiplex means for outputting a combination of outputs from said spectrum parameter calculation means, said adaptive codebook means, said sound source quantization means, and said gain quantization means.

REMARKS

The specification has been amended to correct minor typographical and grammatical errors. No new matter has been added.

Claims 1 to 11 remain in the application. Claims 6, 7 and 8 have been amended.

Claims 1 to 11 were rejected under 35 U.S.C. §112, first paragraph, as containing subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. The Examiner states that claims 1 to 11, as argued by applicant, include subject matter for processing pulses by using different pulse-shifting schemes depending upon whether a voice sound mode or an unvoiced sound mode is discriminated. The Examiner contends that the specification does not support a time shift of both voiced and unvoiced sound modes. The rejection is respectfully traversed.

The Examiner refers to page 29, lines 14 to 19, of the specification, saying that the time shift amount is indicated by $\delta(j)$. It appears that the Examiner's page